Amendments to the Claims:

This listing of claims replaces all prior versions and listings of claims in the application:

Listing of Claims:

1. (Currently Amended) A method for temporal drift correction in a real-time electronic communication comprising:

measuring a size of a receiving data buffer;

comparing the measured size of the receiving data buffer to a predetermined nominal data buffer size to produce a comparison result;

weighting the comparison result[[;]] with ~~determining~~ a parameter that relates to and amplifies a perceived value of [[the]] temporal drift ~~based on the weighted comparison result~~;

determining, based on the comparison result weighted with the ~~determined~~ parameter, a number of samples to be inserted in or removed from a playback data block; and

modifying the playback data block by inserting or removing a number of samples that is based on the determined number of samples.

2. (Original) The method of claim 1 wherein the number of samples is modified without introducing audible artifacts.

3. (Original) The method of claim 1 wherein measuring the size of the receiving data buffer comprises measuring an instantaneous size of the receiving data buffer.

4. (Original) The method of claim 3 wherein measuring the size of the receiving data buffer comprises:

measuring an instantaneous communication delay associated with the receiving data buffer two or more times; and

averaging the measurements.

5. (Original) The method of claim 1 wherein the real-time electronic communication includes an audio communication.

6. (Original) The method of claim 5 wherein modifying the number of samples comprises performing heuristic resampling of the playback data block.

7. (Original) The method of claim 6 wherein performing heuristic resampling comprises:

analyzing multiple consecutive samples of audio data in the playback data block;

identifying consecutive samples with minimal variation in a parameter of their data; and

adjusting the number of samples in the identified consecutive samples.

8. (Original) The method of claim 7 wherein adjusting the number of samples comprises removing a sample from the identified consecutive samples.

9. (Original) The method of claim 7 wherein adjusting the number of samples comprises adding a sample to the identified consecutive samples.

10. (Currently Amended) A computer program, residing on a computer-readable medium, for correcting temporal drift in a real-time electronic communication, comprising instructions for causing a computer to:

measure a size of a receiving data buffer;

compare the measured size of the receiving data buffer to a predetermined nominal data buffer size to produce a comparison result;

weight the comparison result[[;]] with ~~determine~~ a parameter that relates to and amplifies a perceived value of [[the]] temporal drift ~~based on the weighted comparison result~~;

determine, based on the comparison result weighted with the ~~determined~~ parameter, a number of samples to be inserted in or removed from a playback data block; and

modify the playback data block by inserting or removing a number of samples that is based on the determined number of samples.

11. (Original) The computer program of claim 10 wherein the number of samples is modified without introducing audible artifacts.

12. (Original) The computer program of claim 10 wherein instructions for causing a computer to measure the size of the receiving data buffer comprise instructions for causing a computer to measure an instantaneous size of the receiving data buffer.

13. (Original) The computer program of claim 12 wherein instructions for causing a computer to measure the communication delay comprise instructions for causing a computer to:

measure the instantaneous size of the receiving data buffer two or more times; and

average the measurements.

14. (Original) The computer program of claim 10 wherein the real-time electronic communication includes an audio communication.

15. (Original) The computer program of claim 14 wherein instructions for causing a computer to modify the number of samples comprises instructions for causing a computer to perform heuristic resampling of the playback data block.

16. (Original) The computer program of claim 15 wherein instructions for causing a computer to perform heuristic resampling comprise instructions for causing a computer to:

analyze multiple consecutive samples of audio data in the playback data block;

identify consecutive samples with minimal variation in a parameter of their data;

and adjust the number of samples in the identified consecutive samples.

17. (Currently Amended) A computer system running programmed processes comprising a process for correcting temporal drift in a real-time electronic communication, the process causing the computer system to:

measure a size of a receiving data buffer;

compare the measured size of the receiving data buffer to a predetermined nominal data buffer size to produce a comparison result;

weight the comparison result[[;]] with ~~determine~~ a parameter that relates to and amplifies a perceived value of [[the]] temporal drift ~~based on the weighted comparison result~~;

determine, based on the comparison result weighted with the ~~determined~~ parameter, a number of samples to be inserted in or removed from a playback data block; and

modify the playback data block by inserting or removing a number of samples that is based on the determined number of samples.

18. (Original) The computer system of claim 17 wherein the number of samples is modified without introducing audible artifacts.

19. (Original) The computer system of claim 17 wherein measuring the size of the receiving data buffer comprises measuring an instantaneous size of the receiving data buffer.

20. (Original) The computer system of claim 19 wherein measuring the size of the receiving data buffer comprises:

measuring the instantaneous communication delay associated with the receiving data buffer two or more times; and

averaging the measurements.

21. (Original) The computer system of claim 17 wherein the real-time electronic communication includes an audio communication.

22. (Original) The computer system of claim 21 wherein modifying the number of samples comprises performing heuristic resampling of the audio playback data block.

23. (Original) The computer system of claim 22 wherein performing heuristic resampling comprises:

analyzing multiple consecutive samples of audio data in the playback data block;

identifying consecutive samples with minimal variation in a parameter of their data; and

adjusting the number of samples in the identified consecutive sample.

24. (Previously Presented) The method of claim 1 wherein the samples are not associated with a timestamp.

25. (Previously Presented) The computer program of claim 10 wherein the samples are not associated with a timestamp.

26. (Previously Presented) The computer system of claim 17 wherein the samples are not associated with a timestamp.

27. (New) A method for temporal drift correction in a real-time electronic communication, comprising:

comparing a measured size of a receiving data buffer to a predetermined nominal data buffer size to produce a comparison result;

weighting the comparison result with a parameter that relates to and amplifies a perceived value of temporal drift;

dividing the weighted comparison result by a number of nominal playback blocks in the data buffer; and

determining, based on the comparison result weighted with the parameter and divided by the number of nominal playback blocks, a number of samples to be inserted in or removed from a playback data block,

wherein:

the comparing, weighting and determining are performed in accordance with the formula $TD[i] = CF * (AS[i] - Ns) / Nb$, where

$TD[i]$ is the perceived value of temporal drift for an i-th playback block,

CF is the parameter and is greater than one,

$AS[i]$ is the measured size of the receiving data buffer for the i-th playback block,

Ns is the predetermined nominal data buffer size, and

Nb is a number of nominal playback blocks in the data buffer.

28. (New) The method of claim 27 wherein the number of samples is modified without introducing audible artifacts.

29. (New) The method of claim 27 wherein measuring the size of the receiving data buffer comprises measuring an instantaneous size of the receiving data buffer.

30. (New) The method of claim 29 wherein measuring the size of the receiving data buffer comprises:

measuring an instantaneous communication delay associated with the receiving data buffer two or more times; and

averaging the measurements.

31. (New) The method of claim 27 wherein the real-time electronic communication includes an audio communication.

32. (New) The method of claim 31 wherein modifying the number of samples comprises performing heuristic resampling of the playback data block.

33. (New) The method of claim 32 wherein performing heuristic resampling comprises:
analyzing multiple consecutive samples of audio data in the playback data block;
identifying consecutive samples with minimal variation in a parameter of their data; and
adjusting the number of samples in the identified consecutive samples.

34. (New) The method of claim 33 wherein adjusting the number of samples comprises removing a sample from the identified consecutive samples.

35. (New) The method of claim 33 wherein adjusting the number of samples comprises adding a sample to the identified consecutive samples.